IN THE CLAIMS

Please amend the claims as follows.

1. (Currently Amended) A system for demodulation of secondary audio program information, said system comprising:

a bandpass filter for isolating said secondary audio program information from a Broadcast Television Standards Committee (BTSC) format composite audio signal;

a Hilbert filter for producing a copy of said secondary audio program information with a phase shift; ~~and~~

a delay module for delaying said secondary audio program information to produce a delayed copy of said secondary audio program information, wherein said delay module is capable of applying a non-unity delay to said secondary audio program information to produce said delayed copy of said secondary audio program information; and

an FM demodulator for demodulating said secondary audio program information using said non-unity delay and a discrete time index with a combination of said copy of said secondary audio program information with a phase shift and [[a]] said delayed copy of said secondary audio program information to produce an FM demodulated signal.

2. (Canceled)

3. (Original) The system of claim 1, further comprising an automatic gain control for normalizing amplitude of an FM carrier signal at said FM demodulator.

4. (Original) The system of claim 1, further comprising a lowpass filter for eliminating noise from said FM demodulated signal.

5. (Original) The system of claim 1, wherein said bandpass filter comprises a Finite Impulse Response filter.

6. (Original) The system of claim 1, wherein said bandpass filter comprises a 32-tap Finite Impulse Response filter.

7. (Original) The system of claim 1, wherein said Hilbert filter comprises an 11-tap Hilbert filter.

8. (Original) The system of claim 1, wherein said Hilbert filter produces a copy of said secondary audio program information with a 90 degree phase shift.

9. (Original) The system of claim 1, wherein said FM demodulator uses a simplified approximation for simplified demodulation of said secondary audio program information.

10. (Original) The system of claim 1, wherein said FM demodulator produces said FM demodulated signal using I(n)*Q(n-d)-Q(n)*I(n-d), wherein I(n) represents said delayed copy of said secondary audio program information, Q(n) represents said copy of said secondary audio program information with a phase shift, d represents a non-unity delay, and n represents a discrete time index.

11. (Original) The system of claim 10, wherein d is 2.

12. (Currently Amended) A method for demodulation of a digital signal, said method comprising:

isolating desired signal information from [[an]] a Broadcast Television Standards Committee format audio signal;

phase shifting a copy of said desired signal information to produce a phase shifted copy of said desired information;

delaying a copy of said desired signal information using a non-unity delay to produce a delayed copy of said desired signal information; and

FM demodulating said desired signal information using said non-unity delay and a discrete time index with a combination of said phase shifted copy of said desired signal information and said delayed copy of said desired signal information to produce an FM demodulated signal.

13. (Original) The method of claim 12, further comprising normalizing amplitude of an FM carrier signal at said FM demodulator.

14. (Original) The method of claim 12, further comprising eliminating noise from said FM demodulated signal.

15. (Original) The method of claim 12, wherein said phase shifting step produces a copy of said desired signal information with a 90 degree phase shift.

16. (Original) The method of claim 12, wherein said FM demodulation step uses a simplified approximation for easy demodulation of said desired signal information.

17. (Original) The method of claim 12, wherein said FM demodulation produces said FM demodulated signal using I(n)*Q(n-d)-Q(n)*I(n-d), wherein I(n) represents said delayed copy of said desired signal information, Q(n)

represents said copy of said desired signal information with a phase shift, d represents a non-unity delay, and n represents a discrete time index.

18. (Original) The method of claim 12, where d is 2.

19. (Currently Amended) A method for simplification of secondary audio program signal demodulation, said method comprising:

using a bandpass filter with a minimal number of coefficients to isolate said secondary audio program signal in a composite audio signal;

using a Hilbert filter with a minimal number of coefficients to produce a signal in phase quadrature; ~~and~~

using a delay module to produce a delayed secondary audio program signal using a delay, wherein said delay module is capable of applying a non-unity delay to said secondary audio program signal to produce said delayed secondary audio program signal; and

using a simple approximation for FM demodulation of said secondary audio program signal based on said delay, a time index, said delayed secondary audio program signal, and said signal in quadrature phase.

20. (Original) The method of claim 19, further comprising using automatic gain control to normalize carrier amplitude in said FM demodulation.

21. (Original) The method of claim 19, further comprising using a lowpass filter with a minimal number of coefficients to eliminate noise in said FM demodulated signal.

22. (Original) The method of claim 19, wherein said simple approximation comprises I(n)*Q(n-d)-Q(n)*I(n-d), wherein I(n) represents a delayed copy of said secondary audio program signal, Q(n) represents a copy of said secondary

audio signal with a phase shift, d represents a non-unity delay, and n represents a discrete time index.